code DTMF signals and to determine the frequency content of an audio tone. The routines are thus used in converting user-entered access codes and command select codes from an audio format to a digital logic format useable by microcomputer **16**. The audio path on bus **54** is fed into the two band splitting filters **50** and **52** which have preset break frequencies as discussed above. The circuitry associated with filters **50** and **52** comprises a pair of zero crossing detectors which provide outputs on leads D7 and D6 to microcomputer **16**. Microcomputer **16** responds to the data via leads D7 and D6. In accordance with the routines illustrated in FIGS. **4** through **6**.

FIG. **4** depicts a control routine which is used to properly interface the actual DTMF subroutine (shown in FIGS. **5** and **6**) with the rest of the software. The routine of FIG. **4** calls the DTMF routine and if a valid digit is returned, continues in the loop. If a valid digit is not returned, control exits. If a valid digit is returned, the routine then continues to call the DTMF routine looking for a valid digit. It continues to loop until a valid digit is not found for 40 milliseconds, at which time it returns to the calling program.

With reference to FIG. **5**, the first part of the DTMF routine is of fixed length for timing purposes. This first part accumulates data from the zero crossing detectors over a 33 millisecond time period. This time period is actually six individual 5.5 millisecond time periods. The routine continually loops through this section, accumulating zero crossing data and storing the data in two 6-byte tables. Also during this section, zero crossing interval measurements are made, and the routine exits if either frequency band is out of tolerance. Also, there is a 2 millisecond debounce time preceding this section which makes time period measurements and exits if the times measured are not within predetermined specifications.

Once the data has been accumulated, it is then processed to determine if it was a valid DTMF tone which caused control to proceed through the routine. First, all of the data in each 6-byte table is added up, and then compared to a table containing standard zero crossing data for DTMF tones. If a match is found for both high and low frequency bands, then each individual 5.5 millisecond data byte is compared with another table that contains standard zero crossing data for DTMF tones of that length. If all of those are found to match, the actual button on the telephone device which was depressed is calculated using the location of the data where the match occurred in the table (i.e, the table is in ascending order of tones). If at any time a match is not found, or if a time period error was determined in the first section, the time used up by the routine is calculated and the error is reported. FIG. **5** depicts the first section of the routine while FIG. **6** depicts the second.

Referring now to FIGS. **7** through **11**, the routine for generating tones is illustrated. These routines comprise a first subroutine BTIME used to initialize the variable and switches for the interrupt routine, and also a second subroutine ETIME, which disables interrupts and sets the digital to analog converter to a steady state. Assuming that a two frequency tone is to be generated, e.g. a DTMF tone, the routine must generate two frequencies referred to in the flow chart as frequency **1** and frequency **2**. Referring to FIG. **7**, the slope of frequency **1** and frequency **2** are first set and then the BTIME subroutine is called. The BTIME subroutine is illustrated in FIG. **8**. It operates by accessing initialization tables and then enabling the tone generation circuitry. When the tone has sounded for a sufficient length of time, either a preselected length of time or until the system has changed, the ETIME routine is called to end the tone generation. This routine is illustrated in FIG. **9**, and proceeds by disabling interrupts and disabling the tone generation circuitry.

Tones are generated using an interrupt service routine which is called every 176 microseconds. and the tone data is sent out via the 6-bits B2–B7 of the D to A converter of tone ring generator **56**. The two frequencies are calculated in software as triangular waves, at the frequencies specified by the slopes. These are added together and sent out via the D to A converter of tone ring generator **56**. The output of tone ring generator **56** is a triangular wave which is filtered by hardware into and approximately sinusoidal wave (depending on the frequency of the two tones being generated). The interrupt routine for generating tones is illustrated in FIGS. **10** and **11**.

While the invention has been described in its preferred embodiment, it is to be understood that the invention is capable of modification without departing from the true scope and spirit of the invention in its broader aspects.

What is claimed as novel is as follows:

1. A programmable telephone interface apparatus for coupling a plurality of telephone devices to a telephone utility network, comprising:

programmable switching means, having a plurality of ports, each port being adapted to support bidirectional telephonic communication, including at least one trunk line port adapted for connection to said telephone utility network and at least two device ports each adapted for connection to a telephone device;

said switching means providing a stored configuration of access codes representing predefined connections between selected ones of said ports; and

input means responsive to signals entered through a telephone device communicating thorugh one of said ports for changing said stored configuration;

wherein said switching means is responsive to access code signals originating from said telephone utility network for selectively connecting said trunk line port to a selected one of said device ports.

2. The apparatus of claim **1** further comprising initializing means for establishing an initial configuration of predefined connections.

3. The apparatus of claim **2** wherein said input means alters said initial configuration in response to said signals entered through said telephone device.

4. The apparatus of claim **1** further comprising means for storing said configuration of predefined connections.

5. The apparatus of claim **1** further comprising timing means and wherein said switching means is responsive to said timing means to change said connections between said ports in accordance with said stored configuration.

6. The apparatus of claim **1** wherein said data input means comprises means for decoding DTMF tone signals.

7. The apparatus of claim **1** wherein said data input means comprises means for decoding pulse dialing tone signals.

8. The apparatus of claim **1** further comprising ring voltage generation means responsive to said program-